11-16-00 A

Practitioner's Docket No. 442-009966-US(PAR) **PATENT**

JC966 U.S. PTO 11/15/00

Preliminary Classification:

Proposed Class:

Subclass:

NOTE: "All applicants are requested to include a preliminary classification on newly filed patent applications. The preliminary classification, preferably class and subclass designations, should be identified in the upper right-hand corner of the letter of transmittal accompanying the application papers, for example 'Proposed Class 2, subclass 129.'" M.P.E.P. § 601, 7th ed.

jc690 U.S. PTO 09/713524 11/15/00

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Box Patent Application
Assistant Commissioner for Patents
Washington, D.C. 20231

NEW APPLICATION TRANSMITTAL

Transmitted herewith for filing is the patent application of

Inventor(s): Beghdad AYAD

WARNING: *37 C.F.R. § 1.41(a)(1) points out:*

"(a) A patent is applied for in the name or names of the actual inventor or inventors.

"(1) The inventorship of a nonprovisional application is that inventorship set forth in the oath or declaration as prescribed by § 1.63, except as provided for in § 1.53(d)(4) and § 1.63(d). If an oath or declaration as prescribed by § 1.63 is not filed during the pendency of a nonprovisional application, the inventorship is that inventorship set forth in the application papers filed pursuant to § 1.53(b), unless a petition under this paragraph accompanied by the fee set forth in § 1.17(i) is filed supplying or changing the name or names of the inventor or inventors."

For (title): A NOISE SUPPRESSOR

CERTIFICATION UNDER 37 C.F.R. § 1.10*
(Express Mail label number is mandatory.)
(Express Mail certification is optional.)

I hereby certify that this New Application Transmittal and the documents referred to as attached therein are being deposited with the United States Postal Service on this date 15 November 2000, in an envelope as "Express Mail Post Office to Addressee," mailing Label Number EL627420966US, addressed to the: Assistant Commissioner for Patents, Washington, D.C. 20231.

June Adams
(type or print name of person mailing paper)

June Adams
Signature of person mailing paper

WARNING: *Certificate of mailing (first class) or facsimile transmission procedures of 37 C.F.R. § 1.8 cannot be used to obtain a date of mailing or transmission for this correspondence.*

****WARNING:*** *Each paper or fee filed by "Express Mail" **must** have the number of the "Express Mail" mailing label placed thereon prior to mailing. 37 C.F.R. § 1.10(b).*
*"Since the filing of correspondence under § 1.10 without the Express Mail mailing label thereon is an oversight that can be avoided by the exercise of reasonable care, requests for waiver of this requirement will **not** be granted on petition." Notice of Oct. 24, 1996, 60 Fed. Reg. 56,439, at 56,442.*

1. Type of Application

This new application is for a(n)

(check one applicable item below)

- [X] Original (nonprovisional)
- [ ] Design
- [ ] Plant

WARNING: *Do not use this transmittal for a completion in the U.S. of an International Application under 35 U.S.C. § 371(c)(4), unless the International Application is being filed as a divisional, continuation or continuation-in-part application.*

WARNING: *Do not use this transmittal for the filing of a provisional application.*

NOTE: If one of the following 3 items apply, then complete and attach ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF A PRIOR U.S. APPLICATION CLAIMED and a NOTIFICATION IN PARENT APPLICATION OF THE FILING OF THIS CONTINUATION APPLICATION.

- [ ] Divisional.
- [ ] Continuation.
- [ ] Continuation-in-part (C-I-P).

2. Benefit of Prior U.S. Application(s) (35 U.S.C. §§ 119(e), 120, or 121)

NOTE: A nonprovisional application may claim an invention disclosed in one or more prior filed copending nonprovisional applications or copending international applications designating the United States of America. In order for a nonprovisional application to claim the benefit of a prior filed copending nonprovisional application or copending international application designating the United States of America, each prior application must name as an inventor at least one inventor named in the later filed nonprovisional application and disclose the named inventor's invention claimed in at least one claim of the later filed nonprovisional application in the manner provided by the first paragraph of 35 U.S.C. § 112. Each prior application must also be:

(i) An international application entitled to a filing date in accordance with PCT Article 11 and designating the United States of America; or

(ii) Complete as set forth in § 1.51(b); or

(iii) Entitled to a filing date as set forth in § 1.53(b) or § 1.53(d) and include the basic filing fee set forth in § 1.16; or

(iv) Entitled to a filing date as set forth in § 1.53(b) and have paid therein the processing and retention fee set forth in § 1.21(l) within the time period set forth in § 1.53(f).

37 C.F.R. § 1.78(a)(1).

NOTE: If the new application being transmitted is a divisional, continuation or a continuation-in-part of a parent case, or where the parent case is an International Application which designated the U.S., or benefit of a prior provisional application is claimed, then check the following item and complete and attach ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION(S) CLAIMED.

WARNING: *If an application claims the benefit of the filing date of an earlier filed application under 35 U.S.C. §§ 120, 121 or 365(c), the 20-year term of that application will be based upon the filing date of the earliest U.S. application that the application makes reference to under 35 U.S.C. §§ 120, 121 or 365(c). (35 U.S.C. § 154(a)(2) does not take into account, for the determination of the patent term, any application on which priority is claimed under 35 U.S.C. §§ 119, 365(a) or 365(b).) For a c-i-p application, applicant should review whether any claim in the patent that will issue is supported by an earlier application and, if not, the applicant should consider canceling the reference to the earlier filed application. The term of a patent is not based on a claim-by-claim approach. See Notice of April 14, 1995, 60 Fed. Reg. 20,195, at 20,205.*

WARNING: When the last day of pendency of a provisional application falls on a Saturday, Sunday, or Federal holiday within the District of Columbia, any nonprovisional application claiming benefit of the provisional application must be filed prior to the Saturday, Sunday, or Federal holiday within the District of Columbia. See 37 C.F.R. § 1.78(a)(3).

☐ The new application being transmitted claims the benefit of prior U.S. application(s). Enclosed are ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION(S) CLAIMED.

3. Papers Enclosed

A. Required for filing date under 37 C.F.R. § 1.53(b) (Regular) or 37 C.F.R. § 1.153 (Design) Application

23 Pages of specification

1 Pages of claims

5 Sheets of drawing

WARNING: DO NOT submit original drawings. A high quality copy of the drawings should be supplied when filing a patent application. The drawings that are submitted to the Office must be on strong, white, smooth, and non-shiny paper and meet the standards according to § 1.84. If corrections to the drawings are necessary, they should be made to the original drawing and a high-quality copy of the corrected original drawing then submitted to the Office. Only one copy is required or desired. For comments on proposed then-new 37 C.F.R. § 1.84, see Notice of March 9, 1988 (1990 O.G. 57-62).

NOTE: "Identifying indicia, if provided, should include the application number or the title of the invention, inventor's name, docket number (if any), and the name and telephone number of a person to call if the Office is unable to match the drawings to the proper application. This information should be placed on the back of each sheet of drawing a minimum distance of 1.5 cm. (5/8 inch) down from the top of the page . . ." 37 C.F.R. § 1.84(c)).

(complete the following, if applicable)

☐ The enclosed drawing(s) are photograph(s), and there is also attached a "PETITION TO ACCEPT PHOTOGRAPH(S) AS DRAWING(S)." 37 C.F.R. § 1.84(b).

☐ formal

☐ informal

B. Other Papers Enclosed

____ Pages of declaration and power of attorney

1 Pages of abstract

____ Other

4. Additional papers enclosed

☐ Amendment to claims

☐ Cancel in this applications claims ____________ before calculating the filing fee. (At least one original independent claim must be retained for filing purposes.)

☐ Add the claims shown on the attached amendment. (Claims added have been numbered consecutively following the highest numbered original claims.)

☐ Preliminary Amendment

☒ Information Disclosure Statement (37 C.F.R. § 1.98)

☒ Form PTO-1449 (PTO/SB/08A and 08B)

☒ Citations

☐ Declaration of Biological Deposit

☐ Submission of "Sequence Listing," computer readable copy and/or amendment pertaining thereto for biotechnology invention containing nucleotide and/or amino acid sequence.

☐ Authorization of Attorney(s) to Accept and Follow Instructions from Representative

☐ Special Comments

☐ Other

5. Declaration or oath (including power of attorney)

NOTE: A newly executed declaration is not required in a continuation or divisional application provided that the prior nonprovisional application contained a declaration as required, the application being filed is by all or fewer than all the inventors named in the prior application, there is no new matter in the application being filed, and a copy of the executed declaration filed in the prior application (showing the signature or an indication thereon that it was signed) is submitted. The copy must be accompanied by a statement requesting deletion of the names of person(s) who are not inventors of the application being filed. If the declaration in the prior application was filed under § 1.47, then a copy of that declaration must be filed accompanied by a copy of the decision granting § 1.47 status or, if a nonsigning person under § 1.47 has subsequently joined in a prior application, then a copy of the subsequently executed declaration must be filed. See 37 C.F.R. §§ 1.63(d)(1)-(3).

NOTE: A declaration filed to complete an application must be executed, identify the specification to which it is directed, identify each inventor by full name including family name and at least one given name, without abbreviation together with any other given name or initial, and the residence, post office address and country or citizenship of each inventor, and state whether the inventor is a sole or joint inventor. 37 C.F.R. § 1.63(a)(1)-(4).

☐ Enclosed

Executed by

(check all applicable boxes)

☐ inventor(s).

☐ legal representative of inventor(s). 37 C.F.R. §§ 1.42 or 1.43.

☐ joint inventor or person showing a proprietary interest on behalf of inventor who refused to sign or cannot be reached.

☐ This is the petition required by 37 C.F.R. § 1.47 and the statement required by 37 C.F.R. § 1.47 is also attached. *See* item 13 below for fee.

☒ Not Enclosed.

NOTE: Where the filing is a completion in the U.S. of an International Application or where the completion of the U.S. application contains subject matter in addition to the International Application, the application may be treated as a continuation or continuation-in-part, as the case may be, utilizing ADDED PAGE FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION CLAIMED.

☒ Application is made by a person authorized under 37 C.F.R. § 1.41(c) on behalf of *all* the above named inventor(s).

(The declaration or oath, along with the surcharge required by 37 C.F.R. § 1.16(e) can be filed subsequently).

☐ Showing that the filing is authorized. *(not required unless called into question. 37 C.F.R. § 1.41(d))*

6. Inventorship Statement

WARNING: *If the named inventors are each not the inventors of all the claims an explanation, including the ownership of the various claims at the time the last claimed invention was made, should be submitted.*

The inventorship for all the claims in this application are:

☐ The same.

or

☐ Not the same. An explanation, including the ownership of the various claims at the time the last claimed invention was made,

☐ is submitted.

☐ will be submitted.

7. Language

NOTE: An application including a signed oath or declaration may be filed in a language other than English. An English translation of the non-English language application and the processing fee of $130.00 required by 37 C.F.R. § 1.17(k) is required to be filed with the application, or within such time as may be set by the Office. 37 C.F.R. § 1.52(d).

☒ English

☐ Non-English

☐ The attached translation includes a statement that the translation is accurate. 37 C.F.R. § 1.52(d).

8. Assignment

☒ An assignment of the invention to Nokia Mobile Phones Ltd.

☐ is attached. A separate ☐ "COVER SHEET FOR ASSIGNMENT (DOCUMENT) ACCOMPANYING NEW PATENT APPLICATION" or ☐ FORM PTO 1595 is also attached.

☒ will follow.

NOTE: "If an assignment is submitted with a new application, send two separate letters-one for the application and one for the assignment." Notice of May 4, 1990 (1114 O.G. 77-78).

WARNING: *A newly executed "CERTIFICATE UNDER 37 C.F.R. § 3.73(b)" must be filed when a continuation-in-part application is filed by an assignee. Notice of April 30, 1993, 1150 O.G. 62-64.*

9. Certified Copy

Certified copy(ies) of application(s)

Country	Appln. No.	Filed
Finland	19992453	15 November 1999
Country	Appln. No.	Filed
Country	Appln. No.	Filed

from which priority is claimed

- [X] is (are) attached.
- [ ] will follow.

NOTE: *The foreign application forming the basis for the claim for priority must be referred to in the oath or declaration. 37 C.F.R. § 1.55(a) and 1.63.*

NOTE: *This item is for any foreign priority for which the application being filed directly relates. If any parent U.S. application or International Application from which this application claims benefit under 35 U.S.C. § 120 is itself entitled to priority from a prior foreign application, then complete item 18 on the ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION(S) CLAIMED.*

10. Fee Calculation (37 C.F.R. § 1.16)

A. [X] Regular application

CLAIMS AS FILED						
	Number filed		Number Extra		Rate	Basic Fee 37 C.F.R. § 1.16(a) $ 710.00
Total Claims (37 C.F.R. § 1.16(c))	9	− 20 =	0	×	$ 18.00	0
Independent Claims (37 C.F.R. § 1.16(b))	2	− 3 =	0	×	$ 80.00	0
Multiple dependent claim(s), if any (37 C.F.R. § 1.16(d))				+	$ 270.00	

- [ ] Amendment cancelling extra claims is enclosed.
- [ ] Amendment deleting multiple-dependencies is enclosed.
- [ ] Fee for extra claims is not being paid at this time.

NOTE: *If the fees for extra claims are not paid on filing they must be paid or the claims cancelled by amendment, prior to the expiration of the time period set for response by the Patent and Trademark Office in any notice of fee deficiency. 37 C.F.R. § 1.16(d).*

Filing Fee Calculation $ 710.00

B. [ ] Design application
($ 320.00 —37 C.F.R. § 1.16(f))

Filing Fee Calculation $______

C. [ ] Plant application
($ 490.00—37 C.F.R. § 1.16(g))

Filing fee calculation $______

11. **Small Entity Statement(s)**

☐ Statement(s) that this is a filing by a small entity under 37 C.F.R. § 1.9 and 1.27 is (are) attached.

WARNING: *"Status as a small entity must be specifically established in each application or patent in which the status is available and desired. Status as a small entity in one application or patent does not affect any other application or patent, including applications or patents which are directly or indirectly dependent upon the application or patent in which the status has been established. The refiling of an application under § 1.53 as a continuation, division, or continuation-in-part (including a continued prosecution application under § 1.53(d)), or the filing of a reissue application requires a new determination as to continued entitlement to small entity status for the continuing or reissue application. A nonprovisional application claiming benefit under 35 U.S.C. § 119(e), 120, 121, or 365(c) of a prior application, or a reissue application may rely on a statement filed in the prior application or in the patent if the nonprovisional application or the reissue application includes a reference to the statement in the prior application or in the patent or includes a copy of the statement in the prior application or in the patent and status as a small entity is still proper and desired. The payment of the small entity basic statutory filing fee will be treated as such a reference for purposes of this section." 37 C.F.R. § 1.28(a)(2).*

WARNING: *"Small entity status must not be established when the person or persons signing the . . . statement can unequivocally make the required self-certification." M.P.E.P., § 509.03, 6th ed., rev. 2, July 1996 (emphasis added).*

(complete the following, if applicable)

☐ Status as a small entity was claimed in prior application

______ / ______________, filed on ______________, from which benefit is being claimed for this application under:

35 U.S.C. § ☐ 119(e),
☐ 120,
☐ 121,
☐ 365(c),

and which status as a small entity is still proper and desired.

☐ A copy of the statement in the prior application is included.

Filing Fee Calculation (50% of **A, B** or **C** above)

$______________

NOTE: Any excess of the full fee paid will be refunded if small entitiy status is established and a refund request are filed within 2 months of the date of timely payment of a full fee. The two-month period is not extendable under § 1.136. 37 C.F.R. § 1.28(a).

12. **Request for International-Type Search** (37 C.F.R. § 1.104(d))

(complete, if applicable)

☐ Please prepare an international-type search report for this application at the time when national examination on the merits takes place.

13. **Fee Payment Being Made at This Time**

☐ Not Enclosed

☐ No filing fee is to be paid at this time.
(This and the surcharge required by 37 C.F.R. § 1.16(e) can be paid subsequently.)

☒ Enclosed

☒ Filing fee $ 710.00

☐ Recording assignment
($40.00; 37 C.F.R. § 1.21(h))
(See attached "COVER SHEET FOR ASSIGNMENT ACCOMPANYING NEW APPLICATION".) $ ______

☐ Petition fee for filing by other than all the inventors or person on behalf of the inventor where inventor refused to sign or cannot be reached
($130.00; 37 C.F.R. §§ 1.47 and 1.17(l)) $ ______

☐ For processing an application with a specification in
a non-English language
($130.00; 37 C.F.R. §§ 1.52(d) and 1.17(k)) $ ______

☐ Processing and retention fee
($130.00; 37 C.F.R. §§ 1.53(d) and 1.21(l)) $ ______

☐ Fee for international-type search report
($40.00; 37 C.F.R. § 1.21(e)) $ ______

NOTE: 37 C.F.R. § 1.21(l) establishes a fee for processing and retaining any application that is abandoned for failing to complete the application pursuant to 37 C.F.R. § 1.53(f) and this, as well as the changes to 37 C.F.R. §§ 1.53 and 1.78(a)(1), indicate that in order to obtain the benefit of a prior U.S. application, either the basic filing fee must be paid, or the processing and retention fee of § 1.21(l) must be paid, within 1 year from notification under § 53(f).

Total fees enclosed $ 710.00

14. **Method of Payment of Fees**

☒ Check in the amount of $ 710.00

☐ Charge Account No. ______ in the amount of $ ______

A duplicate of this transmittal is attached.

NOTE: Fees should be itemized in such a manner that it is clear for which purpose the fees are paid. 37 C.F.R. § 1.22(b).

15. Authorization to Charge Additional Fees

WARNING: *If no fees are to be paid on filing, the following items should not be completed.*

WARNING: *Accurately count claims, especially multiple dependent claims, to avoid unexpected high charges, if extra claim charges are authorized.*

[X] The Commissioner is hereby authorized to charge the following additional fees by this paper and during the entire pendency of this application to Account No. 16-1350 :

[XX] 37 C.F.R. § 1.16(a), (f) or (g) (filing fees)

[XX] 37 C.F.R. § 1.16(b), (c) and (d) (presentation of extra claims)

NOTE: *Because additional fees for excess or multiple dependent claims not paid on filing or on later presentation must only be paid or these claims cancelled by amendment prior to the expiration of the time period set for response by the PTO in any notice of fee deficiency (37 C.F.R. § 1.16(d)), it might be best not to authorize the PTO to charge additional claim fees, except possibly when dealing with amendments after final action.*

[XX] 37 C.F.R. § 1.16(e) (surcharge for filing the basic filing fee and/or declaration on a date later than the filing date of the application)

[XX] 37 C.F.R. § 1.17(a)(1)-(5) (extension fees pursuant to § 1.136(a)).

[XX] 37 C.F.R. § 1.17 (application processing fees)

NOTE: *". . .A written request may be submitted in an application that is an authorization to treat any concurrent or future reply, requiring a petition for an extension of time under this paragraph for its timely submission, as incorporating a petition for extension of time for the appropriate length of time. An authorization to charge all required fees, fees under § 1.17, or all required extension of time fees will be treated as a constructive petition for an extension of time in any concurrent or future reply requiring a petition for an extension of time under this paragraph for its timely submission. Submission of the fee set forth in § 1.17(a) will also be treated as a constructive petition for an extension of time in any concurrent reply requiring a petition for an extension of time under this paragraph for its timely submission." 37 C.F.R. § 1.136(a)(3).*

[ ] 37 C.F.R. § 1.18 (issue fee at or before mailing of Notice of Allowance, pursuant to 37 C.F.R. § 1.311(b))

NOTE: *Where an authorization to charge the issue fee to a deposit account has been filed before the mailing of a Notice of Allowance, the issue fee will be automatically charged to the deposit account at the time of mailing the notice of allowance. 37 C.F.R. § 1.311(b).*

NOTE: *37 C.F.R. § 1.28(b) requires "Notification of any change in status resulting in loss of entitlement to small entity status must be filed in the application . . . prior to paying, or at the time of paying, . . . the issue fee. . . " From the wording of 37 C.F.R. § 1.28(b), (a) notification of change of status must be made even if the fee is paid as "other than a small entity" and (b) no notification is required if the change is to another small entity.*

16. Instructions as to Overpayment

NOTE: ". . . Amounts of twenty-five dollars or less will not be returned unless specifically requested within a reasonable time, nor will the payer be notified of such amounts; amounts over twenty-five dollars may be returned by check or, if requested, by credit to a deposit account." 37 C.F.R. § 1.26(a).

[X] Credit Account No. 16-1350

[ ] Refund

SEND ALL CORRESPONDENCE TO:
Clarence A. Green, Reg. No.: 24,622
PERMAN & GREEN, LLP
425 Post Road
Fairfield, Connecticut 06430

SIGNATURE OF PRACTITIONER

Reg. No. 24,622

Clarence A. Green
(type or print name of attorney)

Tel. No. (203) 259-1800

PERMAN & GREEN, LLP
P.O. Address

Customer No. 2512

425 Post Road, Fairfield, Connecticut 06430

☐ **Incorporation by reference of added pages**

(check the following item if the application in this transmittal claims the benefit of prior U.S. application(s) (including an international application entering the U.S. stage as a continuation, divisional or C-I-P application) and complete and attach the ADDED PAGES FOR NEW APPLICATION TRANSMITTAL WHERE BENEFIT OF PRIOR U.S. APPLICATION(S) CLAIMED)

☐ Plus Added Pages for New Application Transmittal Where Benefit of Prior U.S. Application(s) Claimed

Number of pages added ____________

☐ Plus Added Pages for Papers Referred to in Item 4 Above

Number of pages added ____________

☐ Plus added pages deleting names of inventor(s) named in prior application(s) who is/are no longer inventor(s) of the subject matter claimed in this application.

Number of pages added ____________

☐ Plus "Assignment Cover Letter Accompanying New Application"

Number of pages added ____________

☒ **Statement Where No Further Pages Added**

(if no further pages form a part of this Transmittal, then end this Transmittal with this page and check the following item)

☒ This transmittal ends with this page.

Express mail #
EL627420966US

A NOISE SUPPRESSOR

FIELD OF THE INVENTION

This invention relates to noise suppression and is particularly, but not exclusively, related to noise suppression in a speech signal picked up by a mobile terminal such as a mobile phone.

BACKGROUND OF THE INVENTION

When a communications terminal is used to make a record of or to transmit a speech signal containing speech, it is inevitable that its microphone will pick up environmental or background noise from the environment in which a speaking person is located. The background noise reduces the ability of a listener to hear or understand the speech and in some cases, if the noise level is sufficiently high, prevents the listener from hearing anything other than the background noise. In addition, such background noise may have a negative effect on the performance of digital signal processing systems in the communications terminal or in an associated communications network, such as speech coding or speech recognition. Typically, noise suppression systems are incorporated in communications terminals and communications networks to limit the effect of background noise.

Noise suppression has been well known for a number of years. Many different approaches and methods have been proposed to achieve three main ends:

(i) suppressing the noise significantly while preserving good speech quality;
(ii) rapid convergence to the optimal solution independent of the nature of the processed noise; and
(iii) improving speech intelligibility for very low speech-to-noise (SNR) ratios.

One noise suppression method based on the linear Minimum Mean Squared Error (MMSE) criteria will be described with reference to Figure 1. The method operates

on a noisy speech signal $x(t)$ containing a speech signal $s(t)$ and a noise signal $n(t)$ such that $x(t) = s(t) + n(t)$. The noisy speech signal $x(t)$ is in the time domain. It is converted into a sequence of frames having consecutive frame numbers k using a windowing function. The frames are then each transformed into the frequency domain using a Fast Fourier Transform (FFT) in block 10 so as to produce a sequence of noisy speech frames where noisy speech signal $X(f,k)$ in the frequency domain contains a speech signal $S(f,k)$ and a noise signal $N(f,k)$ such that $X(f,k) = S(f,k) + N(f,k)$. The frames in the frequency domain comprise a number of frequency bins f. In the frequency domain, the MMSE approach involves minimising the following error function:

$$\varepsilon^2(f,k) = \mathrm{E}\left\{\left(S(f,k) - \hat{S}(f,k)\right)\cdot\left(S(f,k) - \hat{S}(f,k)\right)^*\right\} \qquad (1)$$

where $\mathrm{E}\{\cdot\}$ is the expectation operator, (*) denotes complex conjugation and $\hat{S}(f,k)$ represents a linear estimate of the input speech signal. The error $\varepsilon^2(f,k)$ defined by Equation 1 represents the squared difference between the true speech component contained within the noisy speech signal and the estimate of that speech component, $\hat{S}(f,k)$, i.e. the estimate of the noise-free speech component. Thus, minimisation of $\varepsilon^2(f,k)$ is equivalent to obtaining the best possible estimate of the speech component. $\hat{S}(f,k)$ is given by:

$$\hat{S}(f,k) = G(f,k)\cdot X(f,k) \qquad (2)$$

where $G(f,k)$ is a gain coefficient. The corresponding solution of the minimisation of $\varepsilon^2(f,k)$ for each frame takes the form of a computation of the gain coefficient $G(f,k)$ which is multiplied by the associated input frequency bin of that frame to produce the estimated noise-free speech component $\hat{S}(f,k)$. This gain coefficient, known as the frequency domain Wiener filter, is given by the ratio below:

$$G(f,k) = \frac{\mathrm{E}\{S(f,k)\cdot X^{*}(f,k)\}}{\mathrm{E}\{X(f,k)\cdot X^{*}(f,k)\}} \quad (3)$$

The Wiener filter $G(f,k)$, is generated for each frequency bin f of each frame.

The noise-suppressed frames are then transformed back into the time domain in block 14 and then combined together to provide a noise suppressed speech signal $\hat{s}(t)$. Ideally, $\hat{s}(t) = s(t)$.

When deriving the Wiener filter, the MMSE approach is equivalent to the orthogonality principle. This principle stipulates that, for each frequency, the input signal $X(f,k)$ is orthogonal to the error $S(f,k) - \hat{S}(f,k)$. This means that:

$$\mathrm{E}\{(S(f,k) - \hat{S}(f,k))\cdot X^{*}(f,k)\} = 0 \quad (4)$$

Because the estimation process is linear, by estimating the signal component of a noisy signal that contains a signal component and a noise component, an estimate of the noise $\hat{N}(f,k)$ is also effectively obtained. Furthermore, the following orthogonality relationship will also be true:

$$\mathrm{E}\{(N(f,k) - \hat{N}(f,k))\cdot X^{*}(f,k)\} = 0 \quad (5)$$

where $\hat{N}(f,k)$ indicates the noise estimate. It also follows that for every frequency, the following equality applies:

$$S(f,k) - \hat{S}(f,k) = \hat{N}(f,k) - N(f,k) \quad (6)$$

that is, the error associated with the estimate of the noise component $\hat{N}(f,k)$ is the same as the error associated with the estimated noise-free speech component $\hat{S}(f,k)$.

In the remainder of this document, the following notation will be adopted: $P_{UV}(f,k)$ is the cross power spectral density between $U(f,k)$ and $V(f,k)$ ($P_{UV}(f,k) = \mathrm{E}\{U(f,k)\cdot V^*(f,k)\}$). $P_{UU}(f,k)$ is the power spectral density (psd) of $U(f,k)$ ($P_{UU}(f,k) = \mathrm{E}\{U(f,k)\cdot U^*(f,k)\}$).

As a consequence of the above-mentioned orthogonality principle, it is possible to derive an expression for the cross psd $P_{SX}(f,k)$, required in order to compute the Wiener filter described by Equation 3:

$$P_{SX}(f,k) = \mathrm{E}\{(X(f,k) - \hat{N}(f,k))\cdot X^*(f,k)\} \tag{7}$$

Moreover, the cross psd $P_{NX}(f,k)$ is given by:

$$P_{NX}(f,k) = \mathrm{E}\{(X(f,k) - \hat{S}(f,k))\cdot X^*(f,k)\} \tag{8}$$

Having in mind the trivial equality $P_{XX}(f,k) = P_{SX}(f,k) + P_{NX}(f,k)$, Equations 3, 6, 7 and 8 introduce and illustrate an idea of *adaptive calculation* since the Wiener filter ($P_{SX}(f,k)/P_{XX}(f,k)$) in Equation 3 depends on the estimated signal $\hat{S}(f,k)$ (6,7) and (8).

When a minimum is reached, the expression describing the error in Equation 2 takes the following form:

$$\varepsilon^2_{\min}(f,k) = \frac{P_{SS}(f,k)\cdot P_{XX}(f,k) - |P_{SX}(f,k)|^2}{P_{XX}(f,k)} \tag{9}$$

It is evident that minimum error, that is $\varepsilon_{\min}^2(f,k)$, is equal to zero only if the desired signal $S(f,k)$ is completely coherent with the input signal $X(f,k)$ (that is, $P_{NN}(f,k)$ tends to zero). This is desirable. Otherwise, there is an error when applying the Wiener filter. The upper limit of this error is $P_{SS}(f,k)$. This is undesirable. In other words, an error-free result can only be obtained if there is actually no noise in the input signal $X(f,k)$. For any finite noise level, a finite error is obtained. It follows that the worst case error occurs when there is no speech signal $S(f,k)$ in $X(f,k)$.

SUMMARY OF THE INVENTION

According to a first aspect of the invention there is provided a method of suppressing noise in a signal containing noise to provide a noise suppressed signal in which an estimate is made of the noise and an estimate is made of speech together with some noise.

Preferably the signal comprises speech.

Preferably the level of the noise included in the estimate of the speech together with some noise is variable so as to include a desired amount of noise in the noise-suppressed signal.

Preferably the level of the noise provides an acceptable level of context information.

Preferably the level of the noise is below the mask limit of the speech and so is not audible to a listener. Alternatively the level of noise approaches the mask limit of the speech and so some noise context information is left in the signal.

Preferably the method does not suppress noise if the signal to noise ratio is sufficiently high so that the level of noise already provides an acceptable level of context information or is already below the mask limit.

Preferably the estimated noise is power spectral density.

According to a second aspect of the invention there is provided a method of producing a gain coefficient for noise suppression in which a first estimation of the gain coefficient is made adaptively and this first estimation is used to produce a noise estimation which is then used to produce a second estimation of the gain function.

In this respect, the invention provides an important advantage. It effectively eliminates the need for a Voice Activity Detector (VAD) in a noise suppressor implemented according to the invention. A VAD is basically an energy detector. It receives a noisy speech signal, compares the energy of the filtered signal with a predetermined threshold and indicates that speech is present in the received signal whenever the threshold is exceeded. In many speech encoding/decoding systems, particularly in the field of mobile telecommunications, operation of the VAD changes the way in which background noise in a speech signal is processed. Specifically, during periods when no speech is detected, transmission may be cut and so-called "comfort noise" generated at the receiving terminal. Thus use of such discontinuous transmission and voice activity detection schemes may complicate the use of noise suppression and lead to unwanted effects. Elimination of the need for a voice activity detector and the creation of a noise suppression scheme that automatically adapts to changes in noise conditions is therefore highly desirable. Because the invention introduces a method of noise suppression in which an estimate of both speech and background noise is obtained, there is effectively no need to make a decision as to whether an input signal contains speech and noise or just noise. As a result the VAD function becomes redundant.

Preferably the first estimation is used to up-date the estimated noise.

According to other aspects of the invention, there is provided a noise suppressor operating according to the first aspect of the invention, a noise suppressor operating according to the second aspect of the invention, a noise suppressor operating according to the first and the second aspects of the invention, a communications terminal comprising a noise suppressor according to the first and/or second aspects of the invention and a communications network comprising a noise suppressor according to the first and/or second aspects of the invention.

Preferably the communications terminal is mobile. Alternatively, the invention may be used in a network or fixed communications terminal.

According to another aspect of the invention there is provided a method of calculating a Wiener filter in which an estimate is made of speech and background noise and the noise is far enough below the speech so that it is wholly or partially masked below the audible level or perception of a user.

Preferably the method is for noise suppression in the frequency domain. It may comprise calculating the numerator and denominator of a Wiener filter to be used for a noise reduction system. The noise suppression system described in this document is particularly suitable for application in a system comprising a single sensor such as a microphone.

Preferably the filter is a Wiener Filter. Preferably it is based on an estimate of a periodogram comprising a combination of speech and noise. Preferably the method involves continuous up-dating of noise psd.

BRIEF DESCRIPTION OF THE DRAWINGS

An embodiment of the invention will now be described by way of example only with reference to the accompanying drawings in which:

Figure 1 shows a mobile terminal according to the invention;

Figure 2 shows a noise suppressor according to the invention;
Figure 3 shows the frequency and sound level dependent masking effect of the human auditory system
Figure 4 shows a block diagram of an algorithm according to the invention; and
Figure 5 shows a functional block diagram of an algorithm according to the invention.

DETAILED DESCRIPTION

In the following the symbol P generally represents power. Where it is primed, that is P', it represents a periodogram and where it is not primed, that is P, it represents a power spectral density (psd). In accordance with their generally accepted meanings, the term "periodogram" is used to denote an average calculated over a short period and the term power spectral density is used to represent a longer term average.

An embodiment of a mobile terminal 10 comprising a noise suppressor 20 according to the invention will now be described with reference to Figure 1. Figure 1 corresponds to an arrangement of a mobile terminal according to the prior art although such prior art terminals comprise conventional prior art noise suppressors. The mobile terminal and the wireless communications system with which it communicates operate according to the Global System for Mobile telecommunications (GSM) standard.

The mobile terminal 10 comprises a transmitting (speech encoding) branch 12 and a receiving (speech decoding) branch 14. In the transmitting (speech encoding) branch 12, a speech signal is picked up by a microphone 16 and sampled by an analogue-to-digital (A/D) converter 18 and noise suppressed in the noise suppressor 20 to produce an enhanced signal. This requires the spectrum of the background noise to be estimated so that background noise in the sampled signal can be suppressed. A typical noise suppressor operates in the frequency domain. The time domain signal is first transformed into the frequency domain which can

be carried out efficiently using a Fast Fourier Transform (FFT). In the frequency domain, voice activity is distinguished from background noise and when there is no voice activity, the spectrum of the background noise is estimated. Noise suppression gain coefficients are then calculated on the basis of the current input signal spectrum and the background noise estimate. Finally, the signal is transformed back to the time domain using an inverse FFT (IFFT).

The enhanced (noise suppressed) signal is encoded by a speech encoder 22 to extract a set of speech parameters which are then channel encoded in a channel encoder 24, where redundancy is added to the encoded speech signal in order to provide some degree of error protection. The resultant signal is then up-converted into a radio frequency (RF) signal and transmitted by a transmitting/receiving unit 26. The transmitting/receiving unit 26 comprises a duplex filter (not shown) connected to an antenna to enable both transmission and reception to occur.

A noise suppressor suitable for use in the mobile terminal of Figure 1 is described in published document WO97/22116.

In order to lengthen battery life, different kinds of input signal-dependent low power operation modes are typically applied in mobile telecommunication systems. These arrangements are commonly referred to as discontinuous transmission (DTX). The basic idea in DTX is to discontinue the speech encoding/decoding process in non-speech periods. Typically, some kind of comfort noise signal, intended to resemble the background noise at the transmitting end, is produced as a replacement for actual background noise.

The speech encoder 22 is connected to a transmission (TX) DTX handler 28. The TX DTX handler 28 receives an input from a voice activity detector (VAD) 30 which indicates whether there is a voice component in the noise suppressed signal provided as the output of noise suppressor block 20. If speech is detected in a signal, its transmission continues. If speech is not detected, transmission of the noise suppressed signal is stopped until speech is detected again.

In the receiving (speech decoding) branch 14 of the mobile terminal, an RF signal is received by the transmitting/receiving unit 26 and down-converted from RF to base-band signal. The base-band signal is channel decoded by a channel decoder 32. If the channel decoder detects speech in the channel decoded signal, the signal is speech decoded by a speech decoder 34.

The mobile terminal also comprises a bad frame handling unit 38 to handle bad, that is corrupted, frames.

The signal produced by the speech decoder, whether decoded speech, comfort noise or repeated and attenuated frames is converted from digital to analogue form by a digital-to-analogue converter 40 and then played through a speaker or earpiece 42, for example to a listener.

Further details of the noise suppressor 20 are shown in Figure 2. It comprises a Fast Fourier Transform, a gain coefficient or Wiener filter calculation block and an Inverse Fast Fourier Transform. Noise suppression is carried out in the frequency domain by multiplying frames by gain coefficients/Wiener filters.

The operation of the noise suppressor 20 will now be described. According to the invention, rather than attempting to estimate the "true" speech component $S(f,k)$ in a noisy speech signal, a Wiener filter is used to estimate a combination of speech and a certain amount of noise according to the relationship $S(f,k)+\xi\cdot N(f,k)$. The modified Wiener filter thus created takes the form:

$$\begin{aligned} G(f,k) &= \frac{P_{(S+\xi\cdot N)X}(f,k)}{P_{XX}(f,k)} \\ &= \frac{P_{SX}(f,k)+\xi\cdot P_{NX}(f,k)}{P_{SX}(f,k)+P_{NX}(f,k)} \end{aligned} \tag{10}$$

Assuming that the speech and noise component are uncorrelated (that is, the cross psd between the speech and noise components must be equal to zero, $P_{SN}(f,k)=0$), Equation 10 can be re-expressed in the form:

$$G(f,k)=\frac{P_{SS}(f,k)+\xi\cdot P_{NN}(f,k)}{P_{SS}(f,k)+P_{NN}(f,k)} \tag{11}$$

The role of the factor ξ is explained below.

As explained earlier, the main advantage of estimating a combination of speech and a certain amount of noise is that there should be less error associated with the estimation. This benefit becomes further apparent in connection with Equation 12, presented below, which defines the minimum error obtained in this situation:

$$\varepsilon_{\min}^2(f,k)=(1-\xi)^2\cdot\frac{P_{SS}(f,k)\cdot P_{NN}(f,k)}{P_{SS}(f,k)+P_{NN}(f,k)} \tag{12}$$

It can now be understood that as $P_{NN}(f,k)$ tends to zero, equation 12 tends to zero and so the error tends to zero as in the case of the prior art. In common with the prior art, this is desirable. However, since Equation 12 includes the factor of $(1-\xi)^2$ it reaches zero more quickly than in the case of the prior art. On the other hand, as $P_{NN}(f,k)$ increases, $\varepsilon_{\min}^2$ tends to $(1-\xi)^2\cdot P_{SS}(f,k)$. In common with the prior art, this is undesirable. However, the error provided by the method according to the invention is always smaller than that provided by the prior art method described earlier. This advantage arises because the multiplying factor $(1-\xi)^2$ always serves to reduce the amount of error. Furthermore, the factor $(1-\xi)^2$ can be minimised by setting ξ to an appropriate value, in which case the error is further minimised.

In the invention it has been recognised that the value of ξ can be determined to achieve the following results:

1. To provide a value of the product $\xi \cdot P_{NN}(f,k)$ which is "masked" by $P_{SS}(f,k)$. Even though an estimate of combined speech and noise is computed, a listener will hear only speech because the product $\xi \cdot P_{NN}(f,k)$ will be below his audible level of perception. In this way, advantage is taken of the properties of the human auditory system, allowing the speech periodogram to be calculated together with the maximum of masked noise periodogram. When ξ is being applied to achieve this result, it is referred to as ξ_1.

   The "masking" effect is a property of the human auditory system which effectively sets a frequency dependent and sound level dependent lower limit or threshold on auditory perception. Thus, any noise or speech components below the masking threshold will not be perceived (heard) by the listener. It is generally accepted that the masking threshold is approximately 13dB below the current input level, irrespective of frequency. This is illustrated in Figure 3. According to the invention, in order to estimate the pure speech signal (that is, when trying to eliminate all the background noise), it is sufficient to estimate the pure speech signal together with that part of the noise just below the masking threshold.

2. To allow the level for noise reduction at the output to be freely chosen. This can be used to restore near-end context to the signal for the far-end listener. When ξ is being applied to achieve this result, it is referred to as ξ_2. This means that ξ may be chosen in such a way as to ensure adequate noise suppression, but also to permit a certain noise component to remain in the signal at the receiving terminal, such that the background noise appears to naturally represent the background noise present in the environment of a transmitting terminal. In other words it is possible to

choose a value of ξ such that the noise component in a noisy speech signal is not completely eliminated due to the masking effect.

In practical situations, speech signals are non-stationary and therefore require short-term estimation. Thus, instead of using psd functions, as shown in Equation 11, certain terms are replaced with periodograms. Noise may be also non-stationary, but it is generally considered to be stationary, so long-term estimation may be still be used. Hence, the form of the desired Wiener filter is:

$$G(f,k)=\frac{P'_{SS}(f,k)+\xi\cdot P'_{NN}(f,k)}{P'_{SS}(f,k)+P_{NN}(f,k)} \qquad (13)$$

It should be noted that it is also possible to use the background noise power spectral density term $P_{NN}(f,k)$ in the denominator of Equation 13. It should also be appreciated that when $\xi=\xi_1$ is used in Equation 13 above, the term $P'_{SS}(f,k)+\xi_1\cdot P'_{NN}(f,k)$ represents a combination of the speech periodogram and the masked noise periodogram and when $\xi=\xi_2$ is used, the term $P'_{SS}(f,k)+\xi_2\cdot P'_{NN}(f,k)$ represents a combination of the speech periodogram and the permitted noise periodogram. The denominator $P'_{SS}(f,k)+P_{NN}(f,k)$ is composed of the speech periodogram and the noise psd, respectively.

Calculation of the Wiener filter for a current frame k is based on a previous frame $k-1$ as follows. The noise psd $P_{NN}(f,k-1)$, the speech periodogram $P'_{SS}(f,k-1)$ and the number of frames $T(f,k-1)$ for time averaging of previous frames are known. For the current frame k, a combination of the input speech and the noise periodogram $|X(f,k)|^2$ is also known. Rather than $P_{NN}(f,k-1)$, $R_{NN}(f,k-1)$ or $L_{NN}(f,k-1)$ may be used if square root or logarithmic measures are employed, as described later in this description.

An eight-step algorithm is used to calculate the Wiener filter. The eight steps are shown in Figure 4 and are described below.

Step 1: Estimation of a combination of the speech and the noise periodogram $\overline{P}'_{SS}(f,k)$

This periodogram is calculated as follows:

$$\overline{P}'_{SS}(f,k) = \alpha \cdot P'_{SS}(f,k-1) + (1-\alpha) \cdot |X(f,k)|^2 \tag{14}$$

It should be noted that $\overline{P}'_{SS}(f,k)$ is based on the previous periodogram of speech $P'_{SS}(f,k-1)$ and an amount of the current noisy speech signal $|X(f,k)|^2$, determined by a factor α. The value of α is chosen to provide the greatest possible contribution from the current speech component $|S(f,k)|^2$ of the noisy speech SIGNAL $|X(f,k)|^2$, but it is limited to ensure that the factor $(1-\alpha)\cdot|N(f,k)|^2$, which represents the amount of the current noise signal that will be included, is masked by the sum $\alpha \cdot P'_{SS}(f,k-1) + (1-\alpha)\cdot|S(f,k)|^2$ which represents an estimate of the current speech periodogram. Therefore, it should be appreciated that it is necessary to re-calculate the forgetting factor α for every frequency bin f of every frame *k*. It should also be noted that the factor $(1-\alpha)$ referred to in Equation 14 is analogous to ξ_1.

Practically, step 1 is implemented by first estimating the current speech periodogram using the spectral subtraction method described in "*Suppression of Acoustic Noise in Speech Using Spectral Subtraction*", IEEE Trans. On Acoustics Speech and Signal Processing, vol. 27, no. 2, pp. 113-120, April 1979. Then the masking level is set at a value which is approximately 13dB below the estimated speech periodogram level. The noise periodogram is estimated in same way as

the speech periodogram. The value of α is then computed using the mask, the noise periodogram and the input periodogram.

Step 2: Estimation of a combination of speech and noise psd $\overline{P}_{XX}(f,k)$

This psd represents the total power of the input and is estimated by:

$$\overline{P}_{XX}(f,k) = \alpha \cdot \left[P'_{SS}(f,k-1) + \frac{\lambda}{\alpha} P_{NN}(f,k-1) \right] + (1-\alpha) \cdot |X(f,k)|^2 \qquad (15)$$

This psd combines short term averaging (a periodogram for speech) together with long term averaging (a psd for noise).

Step 3: Estimation of the Wiener Filter

The Wiener filter of Equation 11 can be re-written in the following form:

$$G_1(f,k) = \frac{\overline{P}'_{SS}(f,k)}{\overline{P}_{XX}(f,k)} \qquad (16)$$

and so can be calculated from the results of Equations 14 and 15. Since $\hat{S}_1(f,k) = G_1(f,k) \cdot X(f,k)$, it should be understood that the estimated speech $\hat{S}_1(f)$ contains the speech and the masked part of the noise. The minimum value for the gain $G_1(f,k)$ is set to $(1-\alpha)$.

Step 4: Updating of the noise psd $P_{NN}(f,k)$

To update the noise psd, the theoretical result presented in Equation 8 is used, replacing the product $(X(f,k) - \hat{S}(f,k)) \cdot X^*(f,k)$ with the product $(1-G_1(f,k)) \cdot |X(f,k)|^2$ where necessary. The following three methods can be used:

(i) power psd estimation;

(ii) square root psd estimation; and

(iii) logarithm psd estimation.

In all of the methods described below, λ represents a forgetting factor between 0 and 1.

(i) Power psd estimation

This method uses the orthogonality principle and is based on the Welch method described in "The Use of Fast Fourier Transform for the Estimation of Power Spectra: A Method Based on Time Averaging Over Short, Modified Periodograms", IEEE Trans. On Audio and Electroacoustics, vol. AU-15, n. 2, pp. 70-73, June 1967. It uses a technique known as "exponential time averaging", according to which:

$$P_{NN}(f,k) = \lambda \cdot P_{NN}(f,k-1) + (1-\lambda) \cdot (1-G_1(f,k)) \cdot |X(f,k)|^2 \qquad (17)$$

where $G_1(f,k)$ is the Wiener filter calculated according to equation 16.

(ii) Square Root psd estimation

This method uses a modification of the Welch method and is based on amplitude averaging:

$$\begin{cases} R_{NN}(f,k) = \lambda \cdot R_{NN}(f,k-1) + (1-\lambda) \cdot \sqrt{(1-G_1(f,k))} \cdot |X(f,k)| \\ P_{NN}(f,k) = R_{NN}(f,k) \cdot R_{NN}(f,k) \end{cases} \qquad (18)$$

$R_{NN}(f,k)$ represents an average noise amplitude.

(iii) Logarithmic psd estimation

This method uses time averaging in the logarithm domain:

$$\begin{cases} L_{NN}(f,k) = \lambda \cdot L_{NN}(f,k-1) + (1-\lambda) \cdot Log\left[(1-G_1(f,k)) \cdot |X(f,k)|^2\right] \\ P_{NN}(f,k) = \exp[L_{NN}(f,k) + \gamma] \end{cases} \tag{19}$$

$L_{NN}(f,k)$ refers to an average in the logarithmic power domain. γ is Euler's constant and has a value of 0.5772156649.

In each of the three methods described above, the forgetting factor λ plays an important role in the updating of the noise psd and is defined to provide a good psd estimation when noise amplitude is varying rapidly. This is done by relating λ to differences between the current input periodogram $|X(f,k)|^2$ and the noise psd $P_{NN}(f,k-1)$ in the previous frame. λ depends on a value $T(f,k)$ which defines the number of frames used for time averaging and is determined as follows:

$$\begin{cases} if\ |X(f,k)|^2 > 10 \cdot P_{NN}(f,k-1) & T(f,k) = 5 \\ elseif\ |X(f,k)|^2 < 0.1 \cdot P_{NN}(f,k-1) & T(f,k) = 5 \\ else & T(f,k) = Min[T(f,k-1)+1, \quad 20] \end{cases} \tag{20}$$

and λ is derived from $T(f,k)$ as follows:

$$\lambda = \frac{T(f,k)}{T(f,k)+1} \tag{21}$$

It should be noted that it is necessary to re-calculate the forgetting factor λ for each frame k and for every frequency bin f. Clearly, as λ is required in step 2, it needs to be calculated so that it is available for that step. It should also be appreciated that because the noise psd is updated continuously, this removes the need to have a voice activity detector in the noise suppressor 20.

Step 5: Estimation of Current Speech Periodogram $P'_{SS}(f,k)$

The current speech periodogram $P'_{SS}(f,k)$ plays an important role in the algorithm. It is estimated for a current frame so that it can be used in a next frame, that is in Equations 14 and 15. As explained below, $P'_{SS}(f,k)$ should only contain speech and should not contain any noise.

Effectively, after obtaining an estimate of speech amplitude $\hat{S}(f,k)$ in step 3, this step requires estimation of $P'_{SS}(f,k)$ which represents the current speech periodogram.

It is widely accepted that $P'_{SS}(f,k)$ can simply be replaced with the squared estimated speech amplitude, that is: $P'_{SS}(f,k)=\left|\hat{S}(f,k)\right|^2 \textit{estimate of } |S(f,k)|^2$. Unfortunately, a good estimate $\hat{S}(f,k)$ does not actually imply that a good estimate for $|S(f,k)|^2$ can be obtained by simply taking the square. Thus, the method according to the invention seeks to obtain a more accurate estimate $P'_{SS}(f,k)$ of $|S(f,k)|^2$ by applying the MMSE criterion.

Examining the combined speech and noise periodogram, it can be seen that:

$$Y(f,k)=|X(f,k)|^2=|S(f,k)|^2+|N(f,k)|^2+S^*(f,k)\cdot N(f,k)+S(f,k)\cdot N^*(f,k).$$

Thus a good estimate of $|S(f,k)|^2$ may be obtained by minimising the following error (MMSE criterion):

$$\chi^2(f,k)=\mathrm{E}\left\{\left\||S(f,k)|^2-H(f,k)\cdot Y(f,k)\right|^2\right\} \qquad (22)$$

where $H(f,k)\cdot|X(f,k)|^2$ represents an estimate of the speech periodogram $|S(f,k)|^2$.

Direct solution of Equation 22 requires solution of higher order equations, but the solution can be simplified by assuming that the speech and noise are Gaussian processes, uncorrelated with zero means, to provide an approximation of the corresponding Higher Order Wiener filter $H(f,k)$. The approximation used in this method is presented in Equation 23 below. (It should be appreciated that different approximations may be used at this stage without departing from the essential features of the inventive principle).

$$H(f,k)=\frac{3\cdot SNR(f,k)\cdot SNR(f,k)+SNR(f,k)}{3\cdot SNR(f,k)\cdot SNR(f,k)+6\cdot SNR(f,k)+3} \quad (23)$$

Here, $SNR(f,k)$ refers to the signal-to-noise ratio and is calculated as follows:

$$SNR(f,k)=\frac{G_1(f,k)}{1-G_1(f,k)} \quad (24)$$

Equation 24 is the reciprocal of a well-known function relating the Wiener filter and the signal-to-noise ratio. (Wiener = SNR/(SNR+1))

Consequently, the speech periodogram is calculated as follows:

$$P'_{SS}(f,k)=H(f,k)\cdot|X(f,k)|^2 \quad (25)$$

Step 6: The Amplification Function

In conditions of high SNR, when the speech component of the noisy input signal is large compared with the noise component, the estimated Wiener filter $G_1(f,k)$ tends to 1. Furthermore, when the speech to noise ratio is high, $G_1(f,k)$ can be estimated comparatively accurately. Thus, there is a good degree of certainty that the Wiener filter determined in Step 3, offers optimal filtering and provides an output containing a highly accurate estimate of the speech $\hat{S}_1(f)$ with a residual

amount of (masked) noise. As the gain of the filter is close to 1 in this situation, it is advantageous to provide a small amount amplification to bring the gain still closer to 1. However, the additional amplification should also be limited to ensure that Wiener filter gain does not exceed 1 in any circumstance.

On the other hand in conditions where the speech component in the noisy input signal is small compared with the noise component, the opposite is true. The Wiener filter gain is small, and it is likely that $G_1(f,k)$ cannot be determined as accurately as in conditions of high SNR. In this situation, it is not so advantageous to amplify the Wiener filter output and the estimated Wiener filter should be maintained in the form it was originally estimated in step 3.

To take into account these two contradictory requirements that exist in different SNR conditions, the Wiener filter determined in step 3 is modified according to:

$$G_a(f,k) = G_1(f,k)^{Min[Kb(f),\,1-G_1(f,k)]} \qquad (26)$$

to produce a Wiener filter $G_a(f,k)$ to be used in estimation of the final output. $G_a(f,k)$ is a function of $G_1(f,k)$.

Equation 26 exploits the fact that a function such as $y = x^{1-x}$ $(x > 0)$ provides amplification when x is less than one. It therefore fulfils the requirement of providing more amplification in good SNR conditions and less amplification in conditions of low SNR.

The variable $Kb(f)$ can take values between 0 and 1 and is included in the exponent of Equation 26 in order to enable the use of different (e.g. predetermined) amplification levels for different frequency bands *f*, if desired.

Step 7: Selection of the Level of Noise Reduction

In this step, the desired level of noise reduction is selected. For the Wiener filter given in Equation 11, the corresponding ideal temporal output has the form $\hat{s}(t) = s(t) + \xi \cdot n(t)$. Recalling that the noisy input signal has the form $x(t) = s(t) + n(t)$, the noise reduction provided by the filter is theoretically about $20 \cdot \log[\xi]$ dB. This result can be justified by considering the ratio of the noise level in the input signal to that in the output signal (i.e. the signal obtained after noise suppression). This ratio is simply $\xi \cdot n(t) \ / \ n(t)$, which, when expressed as a power ratio in decibels, becomes $20 \cdot \log[\xi]$ dB. Consequently, the factor $0 < \xi < 1$ corresponds to the noise reduction introduced by the filter.

Having chosen a desired noise reduction level and determined the value of ξ necessary to achieve that noise reduction (e.g. for –12 dB noise reduction, $\xi = 0.25$), a factor η is determined such that:

$$G_1(f,k) + \eta \cdot (1 - G_1(f,k)) \Leftrightarrow \frac{P_s(f,k) + \xi \cdot P_n(f,k)}{P_s(f,k) + P_n(f,k)}. \tag{27}$$

Equation 27 presents a way of relating a Wiener filter optimised to provide an output that includes only masked noise to a Wiener filter that provides an output including a certain amount of permitted noise. According to steps 1 - 3, the Wiener filter $G_1(f,k)$ is constructed so as to provide an estimate of the speech component of a noisy speech signal plus an amount of noise which is effectively masked by the speech component. Thus, in the condition where a certain amount of noise is permitted (desired) in the output, the Wiener filter must be modified accordingly. In Equation 27, $G_1(f,k)$ represents the Wiener filter optimised in step 3 to provide an output that contains speech-masked noise. The term $\frac{P_s(f,k) + \xi \cdot P_n(f,k)}{P_s(f,k) + P_n(f,k)}$ represents a Wiener filter that provides an amount of noise reduction ξ, which produces an output signal containing speech and a desired/permitted amount of noise. The term $\eta \cdot (1 - G_1(f,k))$ thus represents an amount of non-masked noise

and is essentially the difference between $\frac{P_s(f,k)+\xi\cdot P_n(f,k)}{P_s(f,k)+P_n(f,k)}$ and $G_1(f,k)$. Taking into account the fact that $G_1(f,k)$ contains noise at a level of about $(1-\alpha)$ times the noise present in the original noisy speech signal, the following relationship between α, η and ξ is true:

$$1-\alpha+\eta\cdot\alpha \Leftrightarrow \xi \tag{28}$$

Step 8: Estimation of the Final Estimated Wiener Filter

Using Equations 16, 26 and 28, the final Wiener filter $G(f,k)$ to be applied to the input is given by:

$$\begin{cases} if\ \alpha > (1-\xi) & \eta = \dfrac{\alpha+\xi-1}{\alpha} \\ else & \eta = 0 \\ G(f,k) = G_a(f,k) + \eta\cdot(1-G_1(f,k)) & \end{cases} \tag{29}$$

Although η depends on α, and has a different value for each frequency bin f of each frame k, the overall noise reduction level is maintained constant around $20\cdot\log[\xi]$ dB.

Alternatively, steps 1 to 8 could be implemented using formulae involving signal-to-noise ratio formulas. In the detailed implementation of steps 1-8, presented above, the discussion was based on calculations of noise psd functions, speech periodograms and input power (periodogram + psd). However, an alternative representation can be obtained by dividing Equation 11 and/or Equation 13 by the noise psd. This alternative representation requires estimation of a (signal+masked noise)-to-noise ratio, instead of a speech periodogram.

An algorithm 50 embodying the invention is shown in Figure 5. The algorithm 50 is shown divided into a set of steps 52 which are an adaptive process and a set of

steps 54 which are a non-adaptive process. The adaptive process uses a computation of the Wiener filter to re-compute the Wiener filter. Accordingly, the step of the computation of the Wiener filter is common both to the adaptive process and to the non-adaptive process.

This Wiener filter calculation is also suitable for minimising the residual echo in a combined acoustic echo and noise control system including one sensor and one loudspeaker.

While preferred embodiments of the invention have been shown and described, it will be understood that such embodiments are described by way of example only. For example, although the invention is described in a noise suppressor located in the up-link path of a mobile terminal, that is providing noise suppressed signal to a speech encoder, it can equally be present in a noise suppressor in the down-link path of a mobile terminal instead of or in addition to the noise suppressor in the up-link path. In this case it could be acting on a signal being provided by a speech decoder. Furthermore, although the invention is described in a mobile terminal, it can alternatively be present in a noise suppressor in a communications network whether used in relation to a speech encoder or a speech decoder.

Numerous variations, changes and substitutions will occur to those skilled in the art without departing from the scope of the present invention. Accordingly, it is intended that the following claims cover all such equivalents or variations as fall within the spirit and scope of the invention.

Claims

1. A method of suppressing noise in a signal containing noise to provide a noise suppressed signal in which an estimate is made of the noise and an estimate is made of speech together with some noise.

2. A method according to claim 1 in which the signal comprises speech.

3. A method according to claim 1 in which the level of the noise included in the estimate of the speech together with some noise is variable so as to include a desired amount of noise in the noise suppressed signal.

4. A method according to claim 3 in which the level of the noise provides an acceptable level of context information.

5. A method according to claim 1 in which the level of the noise is below the mask limit of the speech and so is not audible to a listener.

6. A method according to claim 1 in which the level of noise approaches the mask limit of the speech and so some noise context information is left in the signal.

7. A method of producing a gain coefficient for noise suppression in which a first estimation of the gain coefficient is made adaptively and this first estimation is used to produce a noise estimation which is then used to produce a second estimation of the gain function.

8. A method according to claim 7 in which the estimated noise is power spectral density.

9. A method according to claim 7 in which the first estimation is used to up-date the estimated noise.

Abstract

A method of suppressing noise in a signal containing speech and noise to provide a noise suppressed speech signal. An estimate is made of the noise and an estimate is made of speech together with some noise. The level of the noise included in the estimate of the speech together with some noise is variable so as to include a desired amount of noise in the noise-suppressed signal.

Figure 4

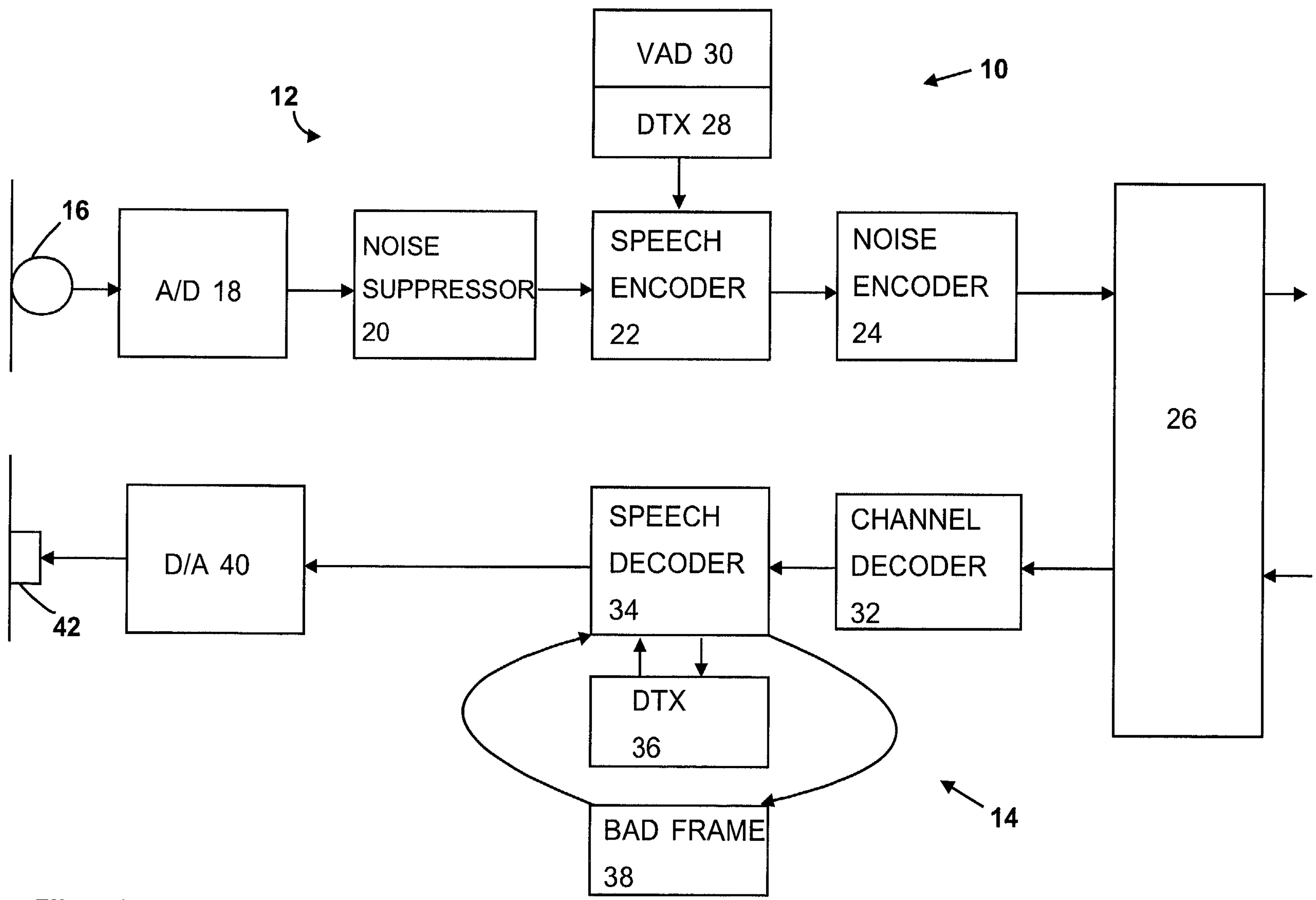
10
12
16
A/D 18
NOISE SUPPRESSOR 20
VAD 30
DTX 28
SPEECH ENCODER 22
NOISE ENCODER 24
26
CHANNEL DECODER 32
SPEECH DECODER 34
DTX 36
BAD FRAME 38
D/A 40
42
14

Fig. 1

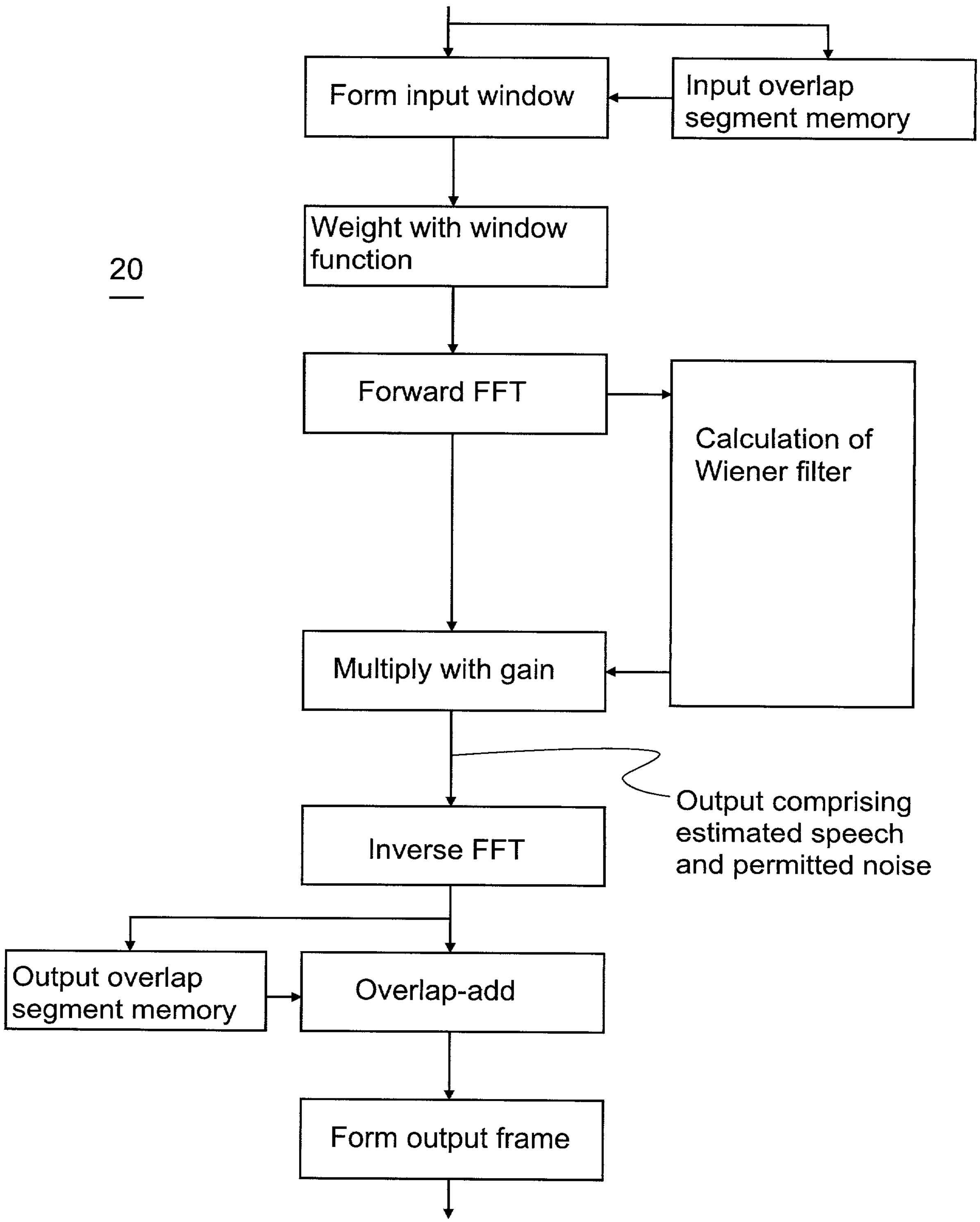
Form input window
Input overlap segment memory
Weight with window function
20
Forward FFT
Calculation of Wiener filter
Multiply with gain
Output comprising estimated speech and permitted noise
Inverse FFT
Output overlap segment memory
Overlap-add
Form output frame

Fig. 2

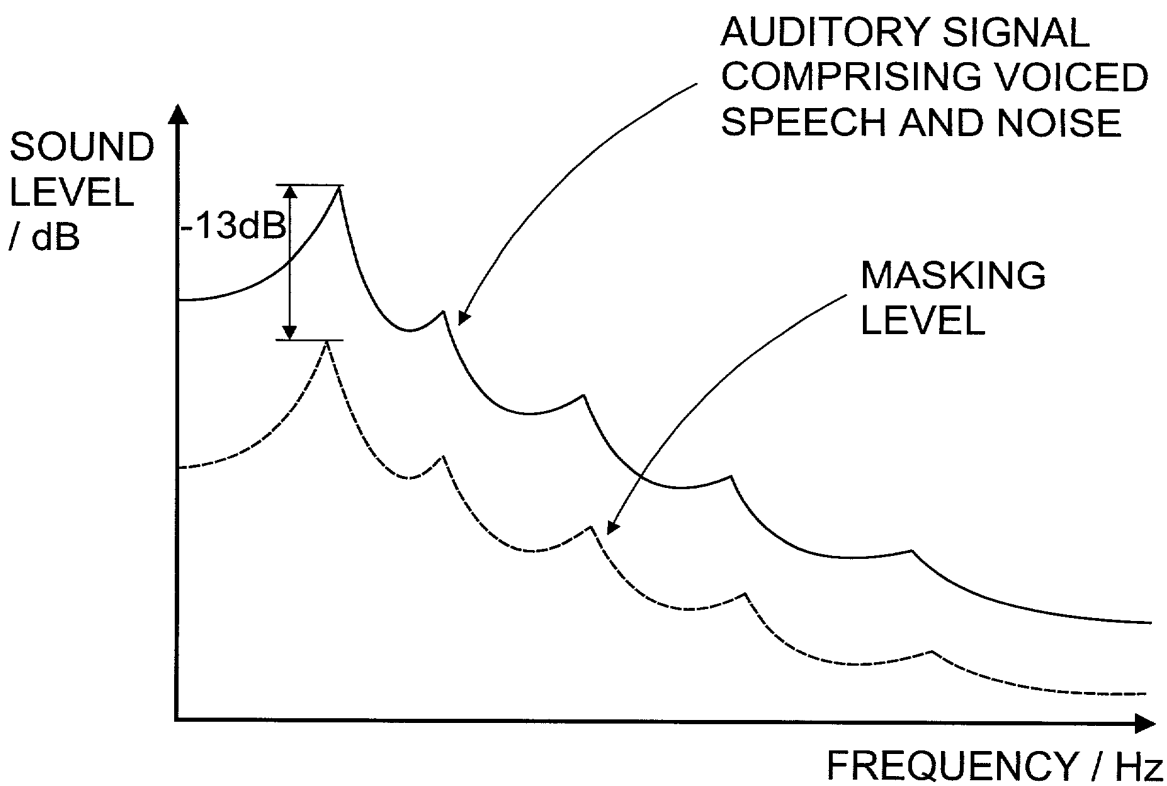
AUDITORY SIGNAL
COMPRISING VOICED
SPEECH AND NOISE
SOUND
LEVEL
/ dB
-13dB
MASKING
LEVEL
FREQUENCY / Hz

Fig. 3

Transform the time domain noisy speech signal input to frequency domain

STEP 1

- Estimate a first speech periodogram
- set the mask at - 13dB of the speech power
- estimate the noise periodogram
- compute the speech+masked noise periodogram
- update the number of block for time averaging
- calculate the forgetting factor for noise psd updating

STEP 2

calculate the input power (speech periodogram + noise psd)

STEP 3

Compute the Wiener filter

STEP 4

update the noise psd

STEP 5

- Estimate the signal-to-noise ratio
- compute the Higher order Wiener filter
- estimate the current speech periodogram

STEP 6

- determine the amplification level at each band
- amplify the Wiener filter

STEP 7

Choose a value for the noise reduction level at the output

STEP 8

compute the final Wiener filter and multiply it with the input to produce the output estimate

Transform the frequency domain estimated output to time domain

Fig. 4

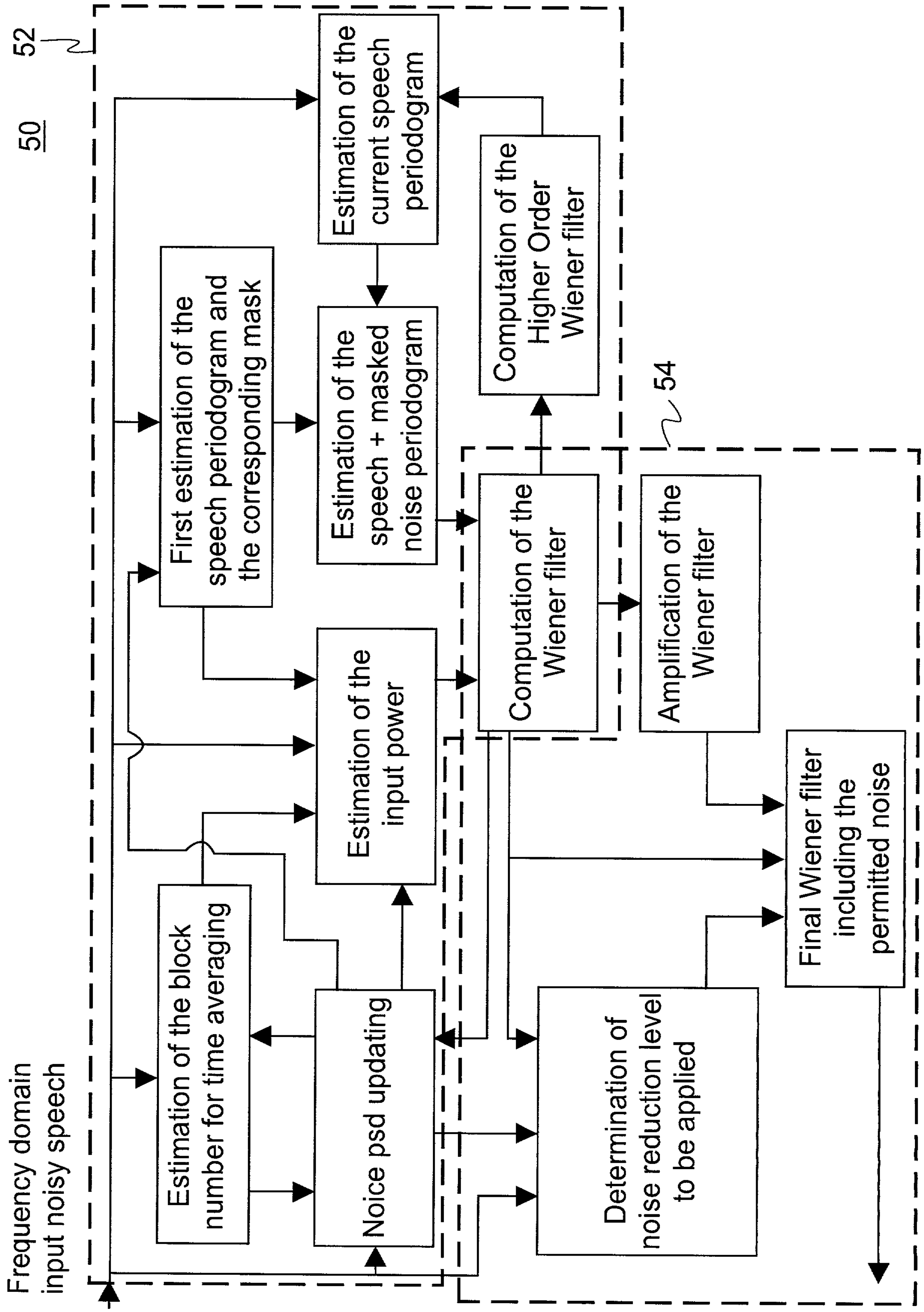
50
52
54
Frequency domain input noisy speech
Estimation of the block number for time averaging
Noice psd updating
Estimation of the input power
First estimation of the speech periodogram and the corresponding mask
Estimation of the speech + masked noise periodogram
Estimation of the current speech periodogram
Computation of the Higher Order Wiener filter
Computation of the Wiener filter
Amplification of the Wiener filter
Determination of noise reduction level to be applied
Final Wiener filter including the permitted noise

Fig. 5